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METHOD AND SYSTEM OF CORRECTING SPECTRAL DEFORMATIONS IN THE VOICE, INTRODUCED BY A COMMUNICATION NETWORK.

BACKGROUND OF THE INVENTION

Field of the invention

The invention concerns a method for the
5 multireference correction of voice spectral
deformations introduced by a communication network. It
also concerns a system for implementing the method.

The aim of the present invention is to improve the
quality of the speech transmitted over communication
10 networks, by offering means for correcting the spectral
deformations of the speech signal, deformations caused
by various links in the network transmission chain.

The description which is given of this hereinafter
explicitly makes reference to the transmission of
15 speech over "conventional" (that is to say cabled)
telephone lines, but also applies to any type of
communication network (fixed, mobile or other)
introducing spectral deformations into the signal, the
parameters taken as a reference for specifying the
20 network having to be modified according to the network.

Description of prior art

The various deformations encountered in the case
of the switched telephone network (STN) will be stated
below.

25 1.1. Degradations in the timbre of the voice on

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the STN network:

Figure 1 depicts a diagram of an STN connection. The speech emitted by a speaker is transmitted by a sending terminal 10, is transported by the subscriber line 20, undergoes an analogue to digital conversion 30 (law A), transmitted by the digital network 40, undergoes a digital (law A) to analogue conversion 50, is transmitted by the subscriber link 60, and passes through the receiving terminal 70 in order finally to be received by the destination person.

Each speaker is connected by an analogue line (twisted pair) to the closest telephone exchange. This is a base band analogue transmission referenced 1 and 3 in Figure 1. The connection between the exchanges follows an entirely digital network. The spectrum of the voice is affected by two types of distortion during the analogue transmission of the base band signal.

The first type of distortion is the bandwidth filtering of the terminals and the points of access to the digital part of the network. The typical characteristics of this filtering are described by UIT-T under the name "intermediate reference system" (IRS) (UIT-T, Recommendation P.48, 1988). These frequency characteristics, resulting from measurements made during the 1970s, are tending however to become obsolete. This is why the UIT-T has recommended since 1996 using a "modified" IRS (UIT-T, Recommendation P.830, 1996), the nominal characteristic of which is depicted in Figure 2 for the transmission part and in Figure 3 for the receiving part. Between 200 and 3400

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Hz, the tolerance is ± 2.5 dB; below
 200 Hz, the decrease in the characteristic of the
 global system must be at least 15 dB per octave. The
 transmission and reception parts of the IRS are called
 5 respectively, according to the UIT-T terminology, the
 "transmitting system" and the "receiving system".

The second distortion affecting the voice spectrum
 is the attenuation of the subscriber lines. In a simple
 model of the local analogue line (given in a CNET
 10 Technical Note NT/LAA/ELR/289 by Cadoret, 1983), it is
 considered that this introduces an attenuation of the
 signal whose value in dB depends on its length and is
 proportional to the square root of the frequency. The
 attenuation is 3 dB at 800 Hz for an average line
 15 (approximately 2 km), 9.5 dB at 800 Hz for longer lines
 (up to 10 km). According to this model, the expression
 for the attenuation of a line, depicted in Figure 4,
 is:

$$20 \quad A_{dB}(f) = A_{dB}(800\text{Hz}) \sqrt{\frac{f}{800}} \quad (0.1)$$

To these distortions there is added the anti-
 aliasing filtering of the MIC coder (ref 30). The
 latter is typically a 200-3400 Hz bandpass filter with
 25 a response which is almost flat over the bandwidth and
 high attenuation outside the band, according to the
 template in Figure 5 for example (National
 Semiconductor, August 1994: Technical Documentation
 TP3054, TP3057).

Finally, the voice suffers spectral distortion as depicted in Figure 6 for the various combinations of three types of analogue line in transmission and reception (that is to say 6 distortions), assuming
 5 equipment complying with the nominal characteristic of the modified SRI. The voice thus appears to be stifled if one of the analogue lines is long and in all cases suffers from a lack of "presence" due to the attenuation of the low-frequency components.

10 1.2. Degradations in the timbre of the voice on the ISDN network and the GSM mobile network

In ISDN and the GSM network, the signal is digitised as from the terminal. The only analogue parts are the transmission and reception transducers
 15 associated with their respective amplification and conditioning chains. The UIT-T has defined frequency efficacy templates for transmission depicted in Figure 7, and for reception depicted in Figure 8, valid both for cabled digital telephones (UIT-T, Recommendation
 20 P.310, May 2000) and mobile digital or wireless terminals (UIT-T, Recommendation P.313, September 1999).

Moreover, for GSM networks, it is recognised that coding and decoding slightly modify the spectral
 25 envelope of the signal. This alteration is shown in Figure 9 for pink noise coded and then decoded in EFR (Enhanced Full Rate) mode.

The effect of these filterings on the timbre is mainly an attenuation of the low-frequency components,
 30 less marked however than in the case of STN.

5 The invention concerns the correction of these spectral distortions by means of a centralized ~~centralised~~ processing, that is to say a device installed in the digital part of the network, as indicated in Figure 10 for the STN.

The objective of a correction of the voice timbre is that the voice timbre in reception is as close as possible to that of the voice emitted by the speaker, which will be termed the original voice.

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2. Prior art

15 Compensation for the spectral distortions introduced into the speech signal by the various elements of the telephone connection is at the present time allowed by devices with an equalization ~~equalisation~~ base. The latter can be fixed or be adapted according to the transmission conditions.

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2.1. Fixed equalization ~~equalisation~~

25 Centralised equalization ~~equalisation~~ devices were proposed in the patents US 5333195 (Duane O. Bowker) and US 5471527 (Helena S. Ho). These equalizers ~~equaliser~~ are fixed filters which restore the level of the low frequencies attenuated by the transmitter. Bowker proposes for example a gain of 10 to 15 dB on the 100-300 Hz band. These methods have two drawbacks:

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* The equalizer ~~equaliser~~ compensates only for the

filtering of the transmitter, so that on reception the low-frequency components remain greatly attenuated by the IRS reception filtering.

5 * This fixed equalization ~~equalisation~~ compensates for the average transmission conditions (transmission system and line). If the actual conditions are too different (for example if the analogue lines are long) the device does not sufficiently correct the timbre, or
10 even impairs it more than the connection without equalization ~~equalisation~~.

2.2. Adaptive equalization ~~equalisation~~

15 The invention described in the patent US 5915235 (Andrew P De Jaco) aims to correct the non-ideal frequency response of a mobile telephone transducer. The equalizer ~~equaliser~~ is described as being placed between the analogue to digital converter and the CELP
20 coder but can be equally well in the terminal or in the network. The principle of equalization ~~equalisation~~ is to bring the spectrum of the received signal close to an ideal spectrum. Two methods are proposed.

25 The first method (illustrated by Figure 4 in the aforementioned patent of De Jaco) consists of calculating long-term autocorrelation coefficients R_{LT} :

$$R_{LT}(n,i) = \alpha R_{LT}(n-1,i) + (1-\alpha)R(n,i), \quad (0.2)$$

30

with $R_{LT}(n,i)$ the i^{th} long-term autocorrelation coefficient to the n^{th} frame, $R(n,i)$ the i^{th} autocorrelation coefficient specific to the n^{th} frame, and α a smoothing constant fixed for example at 0.995.

5 From these coefficients there are derived the long-term LPC coefficients, which are the coefficients of a whitening filter. At the output of this filter, the signal is filtered by a fixed signal which imprints on it the ideal long-term spectral characteristics, i.e.
10 those which it would have at the output of a transducer having the ideal frequency response. These two filters are supplemented by a multiplicative gain equal to the ratio between the long-term energies of the input of the whitener and the output of the second filter.

15

The second method, illustrated by Figure 5 of the aforementioned De Jaco patent, consists of dividing the signal into sub-bands and, for each sub-band, applying a multiplicative gain so as to reach a target energy,
20 this gain being defined as the ratio between the target energy of the sub-band and the long-term energy (obtained by a smoothing of the instantaneous energy) of the signal in this sub-band.

25 These two methods have the drawback of correcting only the non-ideal response of the transmission system and not that of the reception system.

The object of the device of the patent US 5905969
30 (Chafik Mokbel) is to compensate for the filtering of

the transmission signal and of the subscriber line in order to improve the centralised recognition of the speech and/or the quality of the speech transmitted. As presented by Figure 3a in Mokbel, the spectrum of the signal is divided into 24 sub-bands and each sub-band energy is multiplied by an adaptive gain. The matching of the gain is achieved according to the stochastic gradient algorithm, by minimisation of the square error, the error being defined as the difference between the sub-band energy and a reference energy defined for each sub-band. The reference energy is modulated for each frame by the energy of the current frame, so as to respect the natural short-term variations in level of the speech signal. The convergence of the algorithm makes it possible to obtain as an output the 24 equalized ~~equalised~~ sub-band signals.

If the application aimed at is the improvement in the voice quality, the equalized ~~equalised~~ speech signal is obtained by inverse Fourier transform of the equalized ~~equalised~~ sub-band energy.

The Mokbel patent does not mention any results in terms of improvement in the voice quality, and recognises that the method is sub-optimal, in that it uses a circular convolution. Moreover, it is doubtful that a speech signal can be reconstructed correctly by the inverse Fourier transform of band energies distributed according to the MEL scale. Finally, the

device described as not correct the filtering of the reception signal and of the analogue reception line.

The compensation for the line effect is achieved
 5 in the "Mokbel" method of cepstral subtraction, for the purpose of improving the robustness of the speech recognition. It is shown that the cepstrum of the transmission channel can be estimated by means of the mean cepstrum of the signal received, the latter first
 10 being whitened by a pre-accentuation filter. This method affords a clear improvement in the performance of the recognition systems but is considered to be an "off-line" method, 2 to 4 seconds being necessary for estimating the mean cepstrum.

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2.3. Another state of the art combines a fixed pre-equalization ~~pre-equalisation~~ with an adapted equalization ~~equalisation~~ and has been the subject of the filing of a patent application FR 2822999 by the
 20 applicant. The device described aims to correct the timbre of the voice by combining two filters.

A fixed filter, called the pre-equalizer ~~pre-equaliser~~, compensates for the distortions of an
 25 average telephone line, defined as consisting of two average subscriber lines and transmission and reception systems complying with the nominal frequency responses defined in UIT-T, Recommendation P.48, App.I, 1988. Its frequency response on the Fc-3150 Hz band is the
 30 inverse of the global response of the analogue part of

this average connection, F_c being the limit
equalization ~~equalisation~~ low frequency.

5 This pre-equalization ~~pre-equalisation~~ is
supplemented by an adapted equalizer ~~equaliser~~, which
adapts the correction more precisely to the actual
transmission conditions. The frequency response of the
adapted equalizer ~~equaliser~~ is given by:

$$10 \quad |EQ(f)| = \frac{1}{|S_{RX}(f)L_{RX}(f)|} \sqrt{\frac{\gamma_{ref}(f)}{\gamma_x(f)}}, \quad (0.3)$$

with L_{RX} the frequency response of the reception
line, S_{RX} the frequency response of the reception
system and $\gamma_x(f)$ the long-term spectrum of the output x
15 of the pre-equalizer ~~pre-equaliser~~.

The long-term spectrum is defined by the temporal
mean of the short-term spectra of the successive frames
of the signal; $\gamma_{ref}(f)$, referred to as the reference
20 spectrum, is the mean spectrum of the speech defined by
the UIT (UIT-T/P.50/App. I, 1998), taken as an
approximation of the original long-term spectrum of the
speaker. Because of this approximation, the frequency
response of the adapted equalizer ~~equaliser~~ is very
25 irregular and only its general shape is pertinent. This
is why it must be smoothed. The adapted equalizer
~~equaliser~~ being produced in the form of a time filter
RIF, this smoothing in the frequency domain is obtained
by a narrow windowing (symmetrical) of the pulsed

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response.

This method makes it possible to restore a timbre close to that of the original signal on the
5 equalization ~~equalisation~~ band (Fc-3150 Hz), but:

- for some speakers, the approximation of their original long-term spectrum by means of the reference spectrum is very rough, so that the equalizer ~~equaliser~~
10 introduces a perceptible distortion;

- the high smoothing of the frequency response of the equalizer ~~equaliser~~, made necessary by the approximation error, prevents fine spectral distortions
15 from being corrected.

SUMMARY OF THE INVENTION

The aim of the invention is to remedy the
20 drawbacks of the prior art. Its object is a method and system for improving the correction of the timbre by reducing the approximation error in the original long-term spectrum of the speakers.

To this end, it is proposed to classify the
25 speakers according to their long-term spectrum and to approximate this not by a single reference spectrum but by one reference spectrum per class. The method proposed makes it possible to carry out an equalization ~~equalisation~~ processing able to determine the class of
30 the speaker and to equalize ~~equalise~~ according to the

reference spectrum of the class. This reduction in the approximation error makes it possible to smooth the frequency response of the adapted equalizer ~~equaliser~~ less strongly, making it able to correct finer spectral distortions.

The object of the present invention is more particularly a method of correcting spectral deformations in the voice, introduced by a communication network, comprising an operation of equalization ~~equalisation~~ on a frequency band (F1-F2), adapted to the actual distortion of the transmission chain, this operation being performed by means of a digital filter having a frequency response which is a function of the ratio between a reference spectrum and a spectrum corresponding to the long-term spectrum of the voice signal of the speakers, principally characterised in that it comprises:

* prior to the operation of equalization ~~equalisation~~ of the voice signal of a speaker communicating:

- the constitution of classes of speakers with one voice reference per class,

* then, for a given speaker communicating:

- the classification of this speaker, that is to say his allocation to a class from predefined classification criteria in order to make a voice reference which is closest to his own correspond to him,

- the equalization ~~equalisation~~ of the digitised signal of the voice of the speaker carried out with, as

a reference spectrum, the voice reference of the class to which the said speaker has been allocated.

According to another characteristic, the constitution of classes of speakers comprises:

- the choice of a corpus of N speakers recorded under non-degraded conditions and the determination of their long-term frequency spectrum,
- the classification of the speakers in the corpus according to their partial cepstrum, that is to say the cepstrum calculated from the long-term spectrum restricted to the equalization ~~equalisation~~ band (F1-F2) and applying a predefined classification criterion to these cepstra in order to obtain K classes,
- the calculation of the reference spectrum associated with each class so as to obtain a voice reference corresponding to each of the classes.

According to another characteristic, the reference spectrum on the equalization ~~equalisation~~ frequency band (F1-F2), associated with each class, is calculated by Fourier transform of the ~~centre~~ center of the class defined by its partial cepstrum.

According to another characteristic, the classification of a speaker comprises:

- use of the mean pitch of the voice signal and of the partial cepstrum of this signal as classification parameters,
- the application of a discriminating function to these parameters in order to classify the said speaker.

According to the invention the method also

comprises a step of pre-equalization ~~pre-equalisation~~ of the digital signal by a fixed filter having a frequency response in the frequency band (F1-F2), corresponding to the inverse of a reference spectral deformation introduced by the telephone connection.

According to another characteristic, the equalization ~~equalisation~~ of the digitised signal of the voice of a speaker comprises:

- the detection of a voice activity on the line in order to trigger a concatenation of processings comprising the calculation of the long-term spectrum, the classification of the speaker, the calculation of the modulus of the frequency response of the equalizer ~~equaliser~~ filter restricted to the equalization ~~equalisation~~ band (F1-F2) and the calculation of the coefficients of the digital filter differentiated according to the class of the speaker, from this modulus,

- the control of the filter with the coefficients obtained,

- the filtering of the signal emerging from the pre-equalizer ~~pre-equaliser~~ by the said filter.

According to another characteristic, the calculation of the modulus (EQ) of the frequency response of the equalizer ~~equaliser~~ filter restricted to the equalization ~~equalisation~~ band (F1-F2) is achieved by the use of the following equation:

$$|EQ(f)| = \frac{1}{|S_{RX}(f)L_{RX}(f)|} \sqrt{\frac{\gamma_{ref}(f)}{\gamma_x(f)}}, \quad (0.3)$$

in which $\gamma_{\text{ref}}(f)$ is the reference spectrum of the class to which the said speaker belongs,

5 and in which L_{RX} is the frequency response of the reception line, S_{RX} is the frequency response of the reception signal and $\gamma_x(f)$ the long-term spectrum of the input signal x of the filter.

According to a variant, the calculation of the modulus of the frequency response of the equalizer
10 ~~equaliser~~ filter restricted to the equalization ~~equalisation~~ band (F1-F2) is done using the following equation:

$$C_{eq}^p = C_{ref}^p - C_x^p - C_{S_{\text{RX}}}^p - C_{L_{\text{RX}}}^p, \quad (0.13)$$

15

in which C_{eq}^p , C_x^p , $C_{S_{\text{RX}}}^p$ and $C_{L_{\text{RX}}}^p$ are the respective partial cepstra of the adapted equalizer
~~equaliser~~, of the input signal x of the equalizer
~~equaliser~~ filter, of the reception system and of the
20 reception line, C_{ref}^p being the reference partial cepstrum, the ~~centre~~ center of the class of the speaker. The modulus (EQ) restricted to the band F1-F2 is then calculated by discrete Fourier transform of C_{eq}^p .

Another object of the invention is a system for
25 correcting voice spectral deformations introduced by a communication network, comprising adapted equalization
~~equalisation~~ means in a frequency band (F1-F2) which comprise a digital filter whose frequency response is a

function of the ratio between a reference spectrum and a spectrum corresponding to the long-term spectrum of a voice signal, principally characterised in that these means also comprise:

5 - means of processing the signal for calculating the coefficients of the digital signal provided with:

- a signal processing unit for calculating the modulus of the frequency response of the
10 equalizer ~~equaliser~~ filter restricted to the equalization ~~equalisation~~ band (F1-F2) according to the following equation:

$$|EQ(f)| = \frac{1}{|S_{RX}(f)L_{RX}(f)|} \sqrt{\frac{\gamma_{ref}(f)}{\gamma_x(f)}} \quad (0.3)$$

15

in which $\gamma_{ref}(f)$ is the reference spectrum, which may be different from one speaker to another and which corresponds to a reference for a predetermined class to which the said speaker belongs, and in which L_{RX}
20 is the frequency response of the reception line, S_{RX} the frequency response of the reception signal and $\gamma_x(f)$ the long-term spectrum of the input signal x of the filter;

- a second processing unit for calculating the
25 pulsed response from the frequency response modulus thus calculated, in order to determine the coefficients of the filter differentiated according to the class of the speaker.

According to another characteristic, the first processing unit comprises means of calculating the partial cepstrum of the equalizer ~~equaliser~~ filter according to the equation:

5

$$C_{eq}^p = C_{ref}^p - C_x^p - C_{S_RX}^p - C_{L_RX}^p, \quad (0.13)$$

in which C_{eq}^p , C_x^p , $C_{S_RX}^p$ and $C_{L_RX}^p$ are the respective partial cepstra of the adapted equalizer ~~equaliser~~, of the input signal x of the equalizer ~~equaliser~~ filter, of the reception signal and of the reception line, C_{ref}^p being the reference partial cepstrum, the ~~centre~~ center of the class of the speaker, the modulus of (EQ) restricted to the band F1-F2 is then calculated by discrete Fourier transform of C_{eq}^p .

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According to another characteristic, the first processing unit comprises a sub-assembly for calculating the coefficients of the partial cepstrum of a speaker communicating and a second sub-assembly for effecting the classification of this speaker, this second sub-assembly comprising a unit for calculating the pitch F_0 , a unit for estimating the mean pitch from the calculated pitch F_0 , and a classification unit applying a discriminating function to the vector x having as its components the mean pitch and the coefficients of the partial cepstrum for classifying the said speaker.

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According to the invention, the system also

comprises a pre-equalization ~~pre-equalisation~~, the
 signal equalized ~~equalised~~ from reference spectra
 differentiated according to the class of the speaker
 being the output signal x of the pre-equalizer ~~pre-~~
 5 ~~equaliser~~.

BRIEF DESCRIPTION OF THE DRAWINGS

Other particularities and advantages of the
 10 invention will emerge clearly from the following
 description, which is given by way of illustrative and
 non-limiting example and which is made with regard to
 the accompanying figures, which show:

- Figure 1, a diagrammatic telephone connection
 15 for a switched telephone network (STN),
- Figure 2, the transmission frequency response
 curve of the modified intermediate reference system
 IRS,
- Figure 3, the reception frequency response curve
 20 of the modified intermediate reference system IRS,
- Figure 4, the frequency response of the
 subscriber lines according to their length,
- Figure 5, the template of the anti-aliasing
 filter of the MIC coder,
- 25 - Figure 6, the spectral distortions suffered by
 the speech on the switched telephone network with
 average IRS and various combinations of analogue lines,
- Figure 7, the transmission template for the
 digital terminals,
- 30 - Figure 8, the reception template for the digital

terminals,

- Figure 9, the spectral distortion introduced by GSM coding/decoding in EFR (Enhanced Full Rate) mode,

5 - Figure 10, the diagram of a communication network with a system for correcting the speech distortions,

- Figure 11, the steps of calculating the partial cepstrum,

10 - Figure 12, the classification of the partial

cepstra according to the variance criterion,

- Figures 13a and 13b, the long-term spectra corresponding to the ~~centres~~ centers of the classes of speakers respectively for men and women,

15 - Figure 14, the frequency characteristics of the filterings applied to the corpus in order to define the learning corpus,

- Figure 15, the frequency response of the pre-equalizer ~~pre-equaliser~~ for various frequencies F_c ,

20 - Figure 16, the scheme for implementing the system of correction by differentiated equalization ~~equalisation~~ per class of speaker,

- Figure 17, a variant execution of the system according to Figure 16.

25

DETAILED DESCRIPTION OF THE DRAWINGS

Throughout the following the same references entered on the drawings correspond to the same
30 elements.

The description which follows will first of all present the prior step of classification of a corpus of speakers according to their long-term spectrum. This
 5 step defines K classes and one reference per class.

A concatenation of processings makes it possible to process the speech signal (as soon as a voice activity is detected by the system) for each speaker in
 10 order on the one hand to classify the speakers, that is to say to allocate them to a class according to predetermined criteria, and on the other hand to correct the voice using the reference of the class of the speaker.

15

Prior step of classification of the speakers.

* Choice of the class definition corpus.

20 The reference spectrum being an approximation of the original long-term spectrum of the speakers, the definition of the classes of speakers and their respective reference spectra requires having available a corpus of speakers recorded under non-degraded
 25 conditions. In particular, the long-term spectrum of a speaker measured on this recording must be able to be considered to be its original spectrum, i.e. that of its voice at the transmission end of a telephone connection.

30

Definition of the individual: the partial cepstrum

The processing proposed makes it possible to have available, in each class, a reference spectrum as close
 5 as possible to the long-term spectrum of each member of the class. However, only the part of the spectrum included in the ~~equalisation~~ equalization band F1-F2 is taken into account in the adapted ~~equalisation~~ equalization processing. The classes are therefore
 10 formed according to the long-term spectrum restricted to this band.

Moreover, the comparison between two spectra is made at a low spectral resolution level, so as to
 15 reflect only the spectral envelope. This is why the space of the first cepstral coefficients of order greater than 0 (the coefficient of order 0 representing the energy) is preferably used, the choice of the number of coefficients depending on the required
 20 spectral resolution.

The "long-term partial cepstrum", which is denoted C_p , is then determined in the processing as the cepstral representation of the long-term spectrum
 25 restricted to a frequency band. If the frequency indices corresponding respectively to the frequencies F1 and F2 are denoted k_1 and k_2 and the long-term spectrum of the speech is denoted γ , the partial cepstrum is defined by the equation:

30

$$C^p = TFD^{-1}(10\log(\gamma(k_1 \dots k_2) \circ \gamma(k_2 - 1 \dots k_1 + 1))) \quad (0.4)$$

where \circ designates the concatenation operation.

5 The inverse discrete Fourier transform is
calculated for example by IFFT after interpolation of
the samples of the truncated spectrum so as to achieve
a number of power samples of 2. For example, by
choosing the ~~equalisation~~ equalization band
10 187-3187 Hz, corresponding to the frequency indices 5
to 101 for a representation of the spectrum (made
symmetrical) on 256 points (from 0 to 255) the
interpolation is made simply by interposing a frequency
line (interpolated linearly) every three lines in the
15 spectrum restricted to 187-3187 Hz.

The steps of the calculation of the partial
cepstrum are shown in Figure 11.

20 For the cepstral coefficients to reflect the
spectral envelope but not the influence of the harmonic
structure of the spectrum of the speech on the long-
term spectra, the high-order coefficients are not kept.
The speakers to be classified are therefore represented
25 by the coefficients of orders 1 to L of their long-term
partial cepstrum, L typically being equal to 20.

* The classification.

30 The classes are formed for example in a non-

supervised manner, according to an ascending hierarchical classification.

5 This consists of creating, from N separate individuals, a hierarchy of partitionings according to the following process: at each step, the two closest elements are aggregated, an element being either a non-aggregated individual or an aggregate of individuals formed during a previous step. The proximity between
10 two elements is determined by a measurement of dissimilarity which is called distance. The process continues until the whole population is aggregated. The hierarchy of partitionings thus created can be represented in the form of a tree like the one in
15 Figure 12, containing $N-1$ imbricated partitionings. Each cut of the tree supplies a partitioning, which is all the finer, the lower the cut.

20 In this type of classification, as a measurement of distance between two elements, the intra-class inertia variation resulting from their aggregation is chosen. A partitioning is in fact all the better, the more homogeneous are the classes created, that is to say the lower the intra-class inertia. In the case of a
25 cloud of points x_i with respective masses m_i , distributed in q classes with respective ~~centres~~ centers of gravity g_q , the intra-class inertia is defined by:

$$I_{int ra} = \sum_q \sum_{i \in q} m_i \|x_i - g_q\|^2. \quad (0.5)$$

The intra-class inertia, zero at the initial step of the calculation algorithm, inevitably increases with each aggregation.

Use is preferably made of the known principle of aggregation according to variance. According to this principle, at each step of the algorithm used, the two elements are sought whose aggregation produces the lowest increase in intra-class inertia.

The partitioning thus obtained is improved by a procedure of aggregation around the movable ~~centres~~ centers, which reduces the intra-class variance.

The reference spectrum, on the band F1-F2, associated with each class is calculated by Fourier transform of the ~~centre~~ center of the class.

* Example of classification.

The processing described above is applied to a corpus of 63 speakers. The classification tree of the corpus is shown in Figure 12. In this representation, the height of a horizontal segment aggregating two elements is chosen so as to be proportional to their distance, which makes it possible to display the proximity of the elements grouped together in the same

class. This representation facilitates the choice of the level of cutoff of the tree and therefore of the classes adopted. The cutoff must be made above the low-level aggregations, which group together close
 5 individuals, and below the high-level aggregations, which associate clearly distinct groups of individuals.

In this way, four classes are clearly obtained ($K = 4$). These classes are very homogeneous from the point
 10 of view of the sex of the speakers, and a division of the tree into two classes shows approximately one class of men and one class of women.

The consolidation of this partitioning by means of an aggregation procedure around the movable ~~centres~~
 15 centers results in four classes of cardinals 11, 18, 18 and 16, more homogeneous than before from the point of view of the sex: only one man and two women are allocated to classes not corresponding to their sex.

20

The spectra restricted to the 187-3187 Hz band corresponding to the ~~centres~~ centers of these classes are shown in Figures 13a and 13b for the men and women classes as well as for their respective sub-classes.
 25 These spectra, the results of the classification, are used as a multiple reference by the adapted equalizer ~~equaliser~~.

* Use of classification criteria for the speakers
 30

The classes of speakers being defined, the processing provides for the use of parameters and criteria for allocating a speaker to one or other of the classes.

5

This allocation is not carried out simply according to the proximity of the partial cepstrum with one of the class ~~centres~~ centers, since this cepstrum is diverted by the part of the telephone connection upstream of the equalizer ~~equaliser~~.

10

It is advantageously proposed to use classification criteria which are robust to this diversion. This robustness is ensured both by the choice of the classification parameters and by that of the classification criteria learning corpus.

15

* Preferably the classification parameters average pitch and partial cepstrum are used

20

The classes previously defined are homogeneous from the point of view of the sex. The average pitch being both fairly discriminating for a man/woman classification and insensitive to the spectral distortions caused by a telephone connection, and is therefore used as a classification parameter conjointly with the partial cepstrum.

25

* Choice of the classification criteria learning corpus

30

A discrimination technique is applied to these parameters, for example the usual technique of discriminating linear analysis.

5

Other known techniques can be used such as a non-linear technique using a neural network.

If N individuals are available, described by dimension vectors p and distributed a priori in K classes, the discriminating linear analysis consists of:

- firstly, seeking the $K-1$ independent linear functions which best separate the K classes. It is a case of determining which are the linear combinations of the p components of the vectors which minimise the intra-class variance and maximise the inter-class variance;

20

- secondly, determining the class of a new individual by applying the discriminating linear functions to the vector representing him.

In the present case, the vectors representing the individuals have as their components the pitch and the coefficients 1 to L (typically $L = 20$) of the partial cepstrum. The robustness of the discriminating functions to the deviation of the cepstral coefficients is ensured both by the presence of the pitch in the

30

parameters and by the choice of the learning corpus.
The latter is composed of individuals whose original
voice has undergone a great diversity of filtering
representing distortions caused by the telephone
5 connections.

More precisely, from a corpus of original voices
(non-degraded) of N speakers, there is defined a corpus
of N vectors of components $[\bar{F}_0; C^p(I); \dots; C^p(L)]$, with \bar{F}_0 the
10 mean pitch and C^p the partial cepstrum. The construction
of the learning corpus of the said functions consists
of defining a set of M cepstral biases which are each
added to each partial cepstrum representing a speaker
in the original corpus, which makes it possible to
15 obtain a new corpus of NM individuals.

These biases in the domain of the partial cepstrum
correspond to a wide range of spectral distortions of
the band F1-F2, close to those which may result from
20 the telephone connection.

By way of example, the set of frequency responses
depicted in Figure 14 is proposed for the 187-3187 Hz
band: each frequency response corresponds to a path
25 from left to right in the lattice. The amplitude of
their variations on this band does not exceed 20 dB,
like extreme characteristics of the transmission and
line systems.

30 From these 81 frequency characteristics there are

calculated the 81 corresponding biases in the domain of the partial cepstrum, according to the processing described for the use of equation (0.4). By the addition of these biases to the corpus of 63 speakers
 5 previously used, a learning corpus is obtained including 5103 individuals representing various conditions (speaker, filtering of the connection).

In the case of classification by discriminating
 10 linear analysis:

* Application of the classification criteria

Let $(a^k)_{1 \leq k \leq K-1}$ be the family of discriminating
 15 linear functions defined from the learning corpus. A speaker represented by the vector $x = [\overline{F}_0; C^p(I); \dots; C^p(L)]$ is allocated to the class q if the conditional probability of q knowing $a(x)$, denoted $P(q|a(x))$, is maximum, $a(x)$ designating the vector of components $(a^k(x))_{1 \leq k \leq K-1}$.
 20 According to Bayes' theorem,

$$P(q|a(x)) = \frac{P(a(x)|q)P(q)}{P(a(x))}. \quad (0.6)$$

Consequently $P(q|a(x))$ is proportional to
 25 $P(a(x)|q)P(q)$. In the subspace generated by the $K-1$ discriminating functions, on the assumption of a multi-Gaussian distribution of the individuals in each class, the density of probability of $a(x)$ within the class q has:

$$f_q(x) = \frac{1}{(2\pi)^{\frac{K-1}{2}} \sqrt{|S_q|}} \exp \left(-\frac{1}{2} \left(a(x) - a(\bar{x}^q) \right)' S_q^{-1} \left(a(x) - a(\bar{x}^q) \right) \right),$$

(0.7)

5 where \bar{x}^q is the ~~centre~~ center of the class q, $|S_q|$ designates the determinant of the matrix S_q , and S_q is the matrix of the covariances of a within the class q, of generic element σ^{qjk} , which can be estimated by:

$$10 \quad \sigma_{jk}^q = \frac{1}{N_q} \sum_{i=1}^{N_q} \left(a^j(x^i) - a^j(\bar{x}^q) \right) \left(a^k(x^i) - a^k(\bar{x}^q) \right). \quad (0.8)$$

The individual x will be allocated to the class q which maximises $f_q(x)P(q)$, which amounts to minimising on q the function $s_q(x)$ also referred to as the

15 discriminating score:

$$s_q(x) = \left(a(x) - a(\bar{x}^q) \right)' S_q^{-1} \left(a(x) - a(\bar{x}^q) \right) + \log(|S_q|) - 2\log(P(q)),$$

(0.9)

20 The correction method proposed is implemented by the correction system (equalizer ~~equaliser~~) located in the digital network 40 as illustrated in Figure 10.

Figure 16 illustrates the correction system able

25 to implement the method. Figure 17 illustrates this system according to a variant embodiment as will be

detailed hereinafter. These variants relate to the method of calculating the modulus of the frequency response of the adapted equalizer ~~equaliser~~ restricted to the band F1-F2.

5

The pre-equalizer ~~pre-equaliser~~ 200 is a fixed filter whose frequency response, on the band F1-F2, is the inverse of the global response of the analogue part of an average connection as defined previously (UIT-
10 T/P.830, 1996).

The stiffness of the frequency response of this filter implies a long-pulsed response; this is why, so as to limit the delay introduced by the processing, the
15 pre-equalizer ~~pre-equaliser~~ is typically produced in the form of an RII filter, 20th order for example.

Figure 15 shows the typical frequency responses of the pre-equalizer ~~pre-equaliser~~ for three values of F1.
20 The scattering of the group delays is less than 2 ms, so that the resulting phase distortion is not perceptible.

The processing chain 400 which follows allows
25 classification of the speaker and differentiated matched equalization ~~equalisation~~. This chain comprises two processing units 400A and 400B. The unit 400A makes it possible to calculate the modulus of the frequency response of the equalizer ~~equaliser~~ filter restricted
30 to the equalization ~~equalisation~~ band: EQ dB (F1-F2).

The second unit 400B makes it possible to calculate the pulsed response of the equalizer ~~equaliser~~ filter in order to obtain the coefficients
 5 eq(n) of the differentiated filter according to the class of the speaker.

A voice activity frame detector 401 triggers the various processings.

10

The processing unit 410 allows classification of the speaker.

The processing unit 420 calculates the long-term
 15 spectrum followed by the calculation of the partial cepstrum of this speaker.

The output of these two units is applied to the operator 428a or 428b. The output of this operator
 20 supplies the modulus of the frequency response of the equalizer ~~equaliser~~ matched for dB restricted to the equalization ~~equalisation~~ band F1-F2 via the unit 429 for 428a, via the unit 440 for 428b.

25 The processing units 430 to 435 calculate the coefficients eq(n) of the filter.

The output x(n) of the pre-equalizer ~~pre-equaliser~~ is analysed by successive frames with a typical
 30 duration of 32 ms, with an interframe overlap of

typically 50%. For this purpose an analysis window represented by the blocks 402 and 403 is opened.

5 The matched equalization ~~equalisation~~ operation is implemented by an RIF filter 300 whose coefficients are calculated at each voice activity frame by the processing chain illustrated in Figures 16 and 17.

10 The calculation of these coefficients corresponds to the calculation of the pulsed response of the filter from the modulus of the frequency response.

15 The long-term spectrum of $x(n)$, γ_x , is first of all calculated (as from the initial moment of functioning) on a time window increasing from 0 to a voice activity duration T (typically 4 seconds), and then adjusted recursively to each voice activity frame, which is represented by the following generic formula:

20
$$\gamma_x(f,n) = \alpha(n) |X(f,n)|^2 + (1 - \alpha(n)) \gamma_x(f,n-1),$$

(0.10)

25 where $\gamma_x(f,n)$ is the long-term spectrum of x at the n^{th} voice activity frame, $X(f,n)$ the Fourier transform of the n^{th} voice activity frame, and $\alpha(n)$ is defined by equation (0.11). Denoting N the number of frames in the period T ,

$$\alpha(n) = \frac{1}{\min(n, N)} \quad (0.11)$$

This calculation is carried out by the units 421, 422, 423.

5

Next there is calculated, from this long-term spectrum, the partial cepstrum C_p , according to the equation (0.4), used by the processing units 424, 425, 426.

10

The mean pitch \bar{F}_0 is estimated by the processing unit 412 at each voiced frame according to the formula:

$$\bar{F}_0(m) = \alpha(m)F_0(m) + (1 - \alpha(m))\bar{F}_0(m-1), \quad (0.12)$$

15

where $F_0(m)$ is the pitch of the m^{th} voiced frame and is calculated by the unit 411 according to an appropriate method of the prior art (for example the autocorrelation method, with determination of the voicing by comparison of the standardized ~~standardised~~ autocorrelation with a threshold (UIT-T/G.729, 1996)).

20

Thus, at each voice activity frame, there is a new vector x of components, the mean pitch and the coefficients 1 to L of the partial cepstrum, to which there is applied the discriminating function a defined from the learning corpus. This processing is implemented by the unit 413. The speaker is then allocated to the minimum discriminating score class q .

25

The modulus in dB of the frequency response of the matched equalizer ~~equaliser~~ restricted to the band F1-F2, denoted $|EQ|_{dB(F1-F2)}$, is calculated according to one
5 of the following two methods:

The first method (Figure 16) consists of calculating $|EQ|_{F1-F2}$ according to equation (0.3), where $\gamma_{ref}(f)$ is the reference spectrum of the class of the
10 speaker (Fourier transform of the class center ~~centre~~). This calculation method is implemented in this variant depicted in Figure 16 with the operators 414a, 428a, 427 and 429.

The second method (Figure 17) consists of transcribing equation (0.3) into the domain of the partial cepstrum, and then the partial cepstrum of the output x of the pre-equalization ~~pre-equalisation~~,
15 necessary for the classification of the speaker, is available. Thus equation (0.3) becomes:
20

$$C_{eq}^p = C_{ref}^p - C_x^p - C_{S_RX}^p - C_{L_RX}^p, \quad (0.13)$$

where C_{eq}^p , C_x^p , $C_{S_RX}^p$ and $C_{L_RX}^p$ are the respective
25 partial cepstra of the matched equalizer ~~equaliser~~, of the output x of the pre-equalizer ~~pre-equaliser~~, of the reception system and of the reception line, C_{ref}^p being the reference partial cepstrum, the center ~~centre~~ of the class of the speaker. The partial cepstra are

calculated as indicated before, selecting the frequency
band F1-F2. This calculation is made solely for the
coefficients 1 to 20, the following coefficients being
unnecessary since they represent a spectral fineness
5 which will be eliminated subsequently.

The 20 coefficients of the partial cepstrum of the
matched equalizer ~~equaliser~~ are obtained by the
operators 414b and 428b according to equation (0.13).

10

The processing unit 441 supplements these 20
coefficients with zeros, makes them symmetrical and
calculates, from the vector thus formed, the modulus in
dB of the frequency response of the matched equalizer
15 ~~equaliser~~ restricted to the band F1-F2 using the
following equation:

$$EQ_{dB(F_1-F_2)} = TFD^{-1}(C_{eq}^p). \quad (0.14)$$

20 This response is decimated by a factor of 3/4 by the
operator 442.

For the two variants which have just been
described, the values of |EQ| outside the band F1-F2
25 are calculated by linear extrapolation of the value in
dB of |EQ|_{F1-F2}, denoted EQ_{dB} hereinafter, by the unit 430
in the following manner:

For each index of frequency k, the linear
30 approximation of EQ_{dB} is expressed by:

$$EQ_{dB}(k) = \alpha_1 + \alpha_2 k \quad (0.15)$$

The coefficients α_1 and α_2 are chosen so as to
 5 minimise the square error of the approximation on the
 range $F1-F2$, defined by

$$e = \sum_{k=k_1}^{k_2} (EQ_{dB}(k) - EQ_{dB}(k))^2 \quad (0.16)$$

10 The coefficients α_1 and α_2 are therefore defined
 by:

$$\begin{pmatrix} \alpha_1 \\ \alpha_2 \end{pmatrix} = \begin{pmatrix} k_2 - k_1 + 1 \sum_{k=k_1}^{k_2} k \\ \sum_{k=k_1}^{k_2} k & \sum_{k=k_1}^{k_2} k^2 \end{pmatrix}^{-1} \begin{pmatrix} \sum_{k=k_1}^{k_2} EQ_{dB}(k) \\ \sum_{k=k_1}^{k_2} k EQ_{dB}(k) \end{pmatrix} \quad (0.17)$$

15 The values of $|EQ|$, in dB, outside the band $F1-F2$,
 are then calculated from the formula (0.15).

The frequency characteristic thus obtained must be
 smoothed. The filtering being performed in the time
 20 domain, the means allowing this smoothing is to
 multiply by a narrow window the corresponding pulsed
 response.

The pulsed response is obtained by an IFFT
 25 operation applied to $|EQ|$ carried out by the units 431

MARKED-UP SPECIFICATION

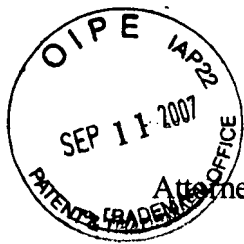
38

and 432 followed by a symmetrization ~~symmetrisation~~
performed by the processing unit 433, so as to obtain a
linear-phase causal filter. The resulting pulsed
response is multiplied, operator 435, by a time window
5 434. The window used is typically a Hamming window of
length 31 ~~centred~~ centered on the peak of the pulsed
response and is applied to the pulsed response by means
of the operator 435.

10

ABSTRACT OF THE DISCLOSURE

5 A technique for correcting the voice spectral
 deformations introduced by a communication network.
 Prior to the operation of equalization ~~equalisation~~ of
 the voice signal of a speaker, the constitution of
 classes of speakers is communicated, with one voice
 10 reference per class. Then, for a given speaker, the
 classification of this speaker is communicated, that is
 to say his allocation to a class from predefined
 classification criteria in order to make a voice
 reference which is closest to his own correspond to
 15 him. Then, for that given speaker, communicating the
equalization ~~equalisation~~ of the digitized ~~digitised~~
 signal of the voice of the speaker carried out with, as
 a reference spectrum, the voice reference of the class
 to which the speaker has been allocated. This technique
 20 applies to the correction of the timbre of the voice in
 switched telephone networks, in ISDN networks and in
 mobile networks.



Attorney Docket # 5394-3

Patent

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In re Application of

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For: Method and System of Correcting Spectral
Deformations in the Voice, Introduced by a
Communication Network

Examiner: Wozniak, James S.
Group Art: 2626

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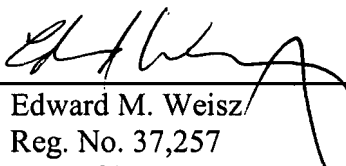
Certification

SIR:

The undersigned attorney certifies that the attached substitute specification does not
contain new matter. All amendments made are clearly shown in the attached redlined
specification.

Respectfully submitted,
COHEN, PONTANI, LIEBERMAN & PAVANE LLP

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